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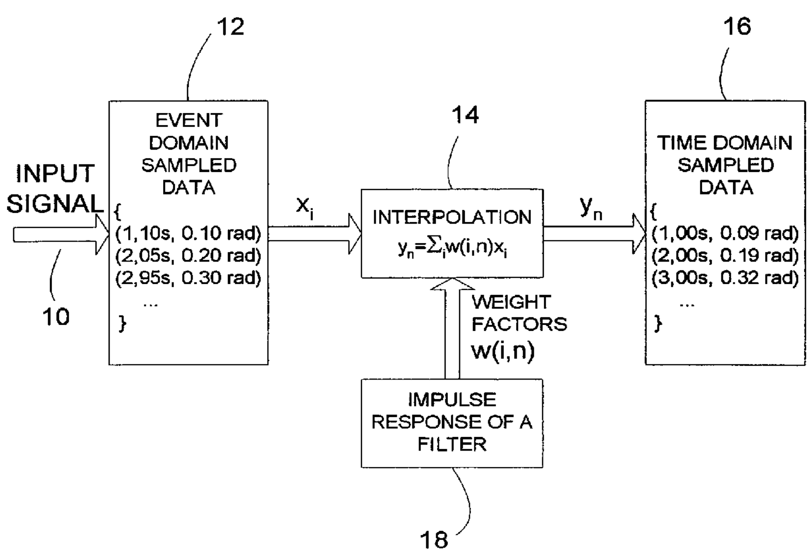
(54) **Title:** METHODS, SYSTEM AND COMPUTER PROGRAM FOR DIGITAL SIGNAL PROCESSING



(57) **Abstract:** The invention is directed to a digital signal processing method and system for digitally processing a discretized in-put signal (12) provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time. The method comprises a conversion of the input signal (12) to a discretized output signal (16) representing the input signal (10) with discrete output signal values at discrete output time instants which are distributed equidistantly in time, wherein the conversion is carried out by a linear interpola-tion (14). The weight factors ($w(i,n)$) of the linear interpo-lation are derived from an impulse response function of a filter (18).

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METHODS, SYSTEM AND COMPUTER PROGRAM FOR DIGITAL SIGNAL PROCESSING

FIELD OF THE INVENTION

The invention relates in general to the field of digital signal processing and more particular to methods, a system and a computer program for processing a discretized input signal provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time.

BACKGROUND OF THE INVENTION

Digital signal processing (DSP) is the processing of signals by digital means. Historically the origins of signal processing are in electrical engineering, and a signal here means an electrical signal carried by a wire or telephone line, or by a radio wave. Following the development of signal processing, a signal nowadays is in general considered a stream of information, for example stock prices, data from a remote-sensing satellite, etc. (for a general introduction to signal processing see the book: Introduction to Signal Processing, S. J. Orfanidis, Prentice Hall 1995).

In many cases, the signal is initially in the form of an analog electrical voltage or current, produced for example by a transducer. An analog signal must be converted into digital (i.e. numerical) form before DSP techniques can be applied. The conversion generates a discretized signal in the form of binary numbers.

In DSP discretized signals are commonly processed in a variety of ways, for example by using a filter circuit in order to remove or at least reduce the unwanted part of the signal. DSP is often implemented in Digital Signal Processor chips - specialized microprocessors with architectures designed specifically for the types of operations required in DSP. DSP technology is nowadays commonplace in such devices as mobile phones, multimedia computers, video recorders, CD players and automobiles.

DSP techniques often comprise Fast Fourier Transforms (FFT), which allow the frequency spectrum of a signal to be rapidly computed. A variety of signal processing methods, software products and microprocessor devices applying such a FFT require equidistantly-sampled input signals. Classical sampling techniques measure the amplitude of a continuous time signal at regular time intervals. In some special applications, the amplitudes of a continuous time signal are however not obtained equidistantly distributed in time. In this case, called event domain sampling, such standard techniques (e.g. FFT, etc.) are not applicable.

In WO 01/87647 A1 which shows the preamble of the claimed subject-matter, such an event domain sampling is disclosed in the context of measuring and analyzing the angular velocity of wheel axles in order to derive an automobile's tire pressure. To be able to apply classical signal processing tools, the document teaches to convert the event domain sampled signal (in form of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time) to a time domain sampled output signal (in form of discrete output signal values at corresponding discrete output time instants which are distributed equidistantly in time, i.e. with a constant sampling period). This conversion applies nearest neighbor linear interpolation. Furthermore, this document discloses a preprocessing of the

measuring signal in order to remove signal noise originating from sensor errors.

OBJECT OF THE INVENTION

The object of the invention is to improve the conversion of event domain sampled input signals to time domain sampled output signal.

SUMMARY OF THE INVENTION

According to a first aspect the invention achieves the object by a digital signal processing method for digitally processing a discretized input signal provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time. The method comprises a conversion of the input signal to a discretized output signal representing the input signal with discrete output signal values at discrete output time instants which are distributed equidistantly in time, wherein the conversion is carried out by a linear interpolation. The weight factors of the linear interpolation are derived from an impulse response function of a filter.

According to a second aspect the invention achieves the object by a digital signal processing method for digitally processing a discretized input signal provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time. The method comprises a conversion of the input signal to a discretized output signal representing the input signal with discrete output signal values at discrete output time instants which are distributed equidistantly in time, wherein the conversion is carried out by a

nearest neighbor linear interpolation. This interpolation is performed with a sampling frequency which is p times higher than the desired output sampling frequency, wherein p is a real number and the output sampling frequency is the reciprocal of the constant differences between the discrete output time instants of the output signal. Subsequently, a digital lowpass filter with a cut-off frequency equal to half of the desired output sampling frequency is applied to the output of the nearest neighbor linear interpolation. Finally, the filter output is decimated by said factor p.

According to a third aspect the invention achieves the object by a digital signal processing system for digitally processing a discretized input signal provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time. The system comprises a conversion component for converting the input signal to a discretized output signal representing the input signal with discrete output signal values at discrete output time instants which are distributed equidistantly in time, wherein the conversion is carried out by a linear interpolation. The weight factors of the linear interpolation are derived from an impulse response function of a filter.

According to a fourth aspect the invention achieves the object by a computer program product including program code for carrying out a digital signal processing method, when executed on a computer system, for digitally processing a discretized input signal provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time. The method comprises a conversion of the input signal to a discretized output signal representing the input signal with discrete output signal values at discrete output time instants which are distributed equidistantly in time, wherein

the conversion is carried out by a linear interpolation. The weight factors of the linear interpolation are derived from an impulse response function of a filter.

Other features inherent in the digital signal processing method and system and the corresponding computer program product are disclosed or will become apparent to those skilled in the art from the following detailed description of embodiments and its accompanying drawings.

DESCRIPTION OF THE DRAWINGS

In the accompanying drawings,

Figs. 1a,b	show two plots of a time domain sampled signal (fig. 1a) and an event domain sampled signal (fig. 1b);
Fig. 2	shows a schematic block diagram illustrating the general method steps as well as the functional means of a preferred embodiment of the invention;
Fig. 3	depicts the Epanechnikov kernel of the linear interpolation method;
Fig. 4	schematically shows a rotational speed sensor for measuring the angular velocity of rotating axles;
Fig. 5	shows a flow diagram of the algorithm for generating of the rotational speed sensor output;
Fig. 6	schematically shows a realistic rotational speed sensor with sensor errors;
Fig. 7	depicts a plot of the rotational velocity of an ideal toothed wheel (dotted line) and of an realistic toothed wheel with sensor errors (solid line);

Fig. 8 shows the Fourier decomposition of an event domain sampled data set corresponding to the measurement of the angular velocity by a non-ideal rotational speed sensor;

Fig. 9 shows a comparison of the Fourier decompositions of a wheel speed sensor's output with and without a preferred signal preprocessing;

Fig. 10 shows a schematic diagram of the preferred signal preprocessing for removing harmonics which overlap with the signal of interest.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

For a better comprehension of the invention, some of the basic terms used for its description are initially explained in more detail with relation to the sampling of a time signal. The invention is however not restricted to such sampled time signals, but may also use any other kind of signal which does not originate from a particular sampling procedure of a time continuous signal but has the same characteristics as described below.

The classical sampling technique measures the amplitude of a continuous time signal $y(t)$ at regular time intervals T_S

$$y_k = y(kT_S)\ , \qquad k = 1,2,\ldots,N\ . \qquad (1)$$

This series y_k is herein referred to as time domain sampled signal.

In event domain sampling the continuous time signal is sampled every time the amplitude of the signal passes certain predefined signal values y_k

$$y(t_k) = y_k \,, \qquad k = 1,2,\ldots,N \,. \tag{2}$$

thus generating a series of data pairs comprising a time instant t_k and a predefined signal value y_k. Besides, equation (2) implies problems when the continuous time signal is not monotonously increasing. However, in many practical applications the continuous time signal is monotonously increasing and the predefined signal values are further uniformly distributed, so that equation (2) can be formulated as

$$y(t_k) = kY_S \,, \qquad k = 1,2,\ldots,N \,, \tag{3}$$

where Y_S is a constant amplitude sample period.

Figs. 1a,b show a continuous time signal 1 sampled in the time domain 2 and the same signal sampled in the event domain 3, respectively. In fig. 1a the time instants 4 at which measurements of the signal are performed, are equidistantly distributed over the time axes 5 with constant sample period T_S. On the signal axes 9 are plotted the signal values y_k according to equation (1). Fig. 1b shows the same time continuous signal, but event domain sampled whereby the predefined signal values y_k are uniformly distributed with sampling period Y_S according to equation (3). It is clearly visible that the thus obtained time intervals are non-equidistantly distributed on the time axes 8.

The constant distance between two time instants in the time domain is called sample distance. The reciprocal of the sample distance is called sampling frequency. Half of the sample frequency is defined as the Nyquist frequency and defines the maximum frequency obtainable in the discretized time domain signal (for an introduction to Fourier decomposition of time domain to frequency domain signal see the book of S. J. Orfanidis). The maximum frequency within the discretized time

domain signal is just half of the Nyquist frequency.

In the following, preferred embodiments of the invention will be explained in general with reference to fig. 2.

According to fig. 2 an input signal 10 is provided in form of an event domain sampled input data set 12 comprising a plurality of discrete input signal values x_i at corresponding discrete input time instants which are distributed non-equidistantly in time. This input data set 12 is converted by an interpolating means 14 to a time domain output data set 16 representing the input signal 10 with discrete output signal values y_n at discrete output time instants which are distributed equidistantly in time.

The input signal 10 may originate from a time continuous signal (for example a measurement signal or a control signal) whereby the term "time continuous" means that a signal value exists for every time instant, hence the signal value is a continuous function of time. When such a time continuous signal will be measured (sampled) by a measuring device, it will be provided in form of the input data set 12. Alternatively, the measurement device may directly output discrete signal values (like the wheel speed sensor described below). The input data set 12 thus may either comprise a list of value pairs of discrete signal values together with corresponding time instant values or, as in the special case of equation (3), a list of discrete time instants only.

The linear interpolation means 14 carry out a linear interpolation of the event domain sampled data 12 and output corresponding time domain sampled data set 16. It uses weight factors for the linear interpolation which are derived from an impulse response function of a filter 18.

In a preferred embodiment, the linear interpolation is carried out according to the following formula:

$$y_n = y\left(t_n^{out}\right) = y\left(nT_{int}\right) = \sum_i w(i,n)x\left(t_i^{in}\right) = \sum_i w(i,n)x_i \ . \qquad (4)$$

Here, T_{int} is the sample period of the output data set 16, $x_i = x\left(t_i^{in}\right)$ denotes the input signal value at the time instant t_i^{in}, $y_n = y\left(t_n^{out}\right)$ denotes the output signal value at the time instant $t_n^{out} = nT_{int}$ and $w(i,n)$ denote the weight factors. The indices i and n denote the i-th element of the input data set 12 respectively the n-th element of the output data set 16. In general, the index i of the sum runs over all elements of the input data set 12. In a specific implementation, however, the range of the index i will be limited in order to reduce computational complexity. In some preferred embodiments, the weight factors are normalized according to $\sum_i w(i,n) = 1$. This will guarantee an unbiased interpolation if $x\left(t_i^{in}\right)$ is constant.

Preferably, the weight factors are chosen according to the following formula so that the linear interpolation is a convolution interpolation:

$$w_h(i,n) = h\left(nT_{\text{int}} - t_i^{in}\right)\left(t_i^{in} - t_{i-1}^{in}\right). \qquad (5)$$

Here, the function $h(t)$ is the impulse response function of a continuous time filter 18. In order to reduce the computational complexity a fixed number M of normalized weights might be preferably used. The output data values are then obtained according to:

$$y(nT_{\mathrm{int}}|t_k) = \frac{\sum_{i=k-M}^{k} w_h(i,n)x(t_i)}{\sum_{i=k-M}^{k} w_h(i,n)} = \sum_{i=k-M}^{k} w(i,n)x(t_i)\ . \tag{6}$$

Limiting the number of weights to M has the consequence that only a fixed number of components of the discretized impulse response of the filter is used. Preferably, the underlying filter is causal, requiring that $t_k < nT_{int}$. The upper limit k of the summation is preferably chosen in such that $nT_{int} - t_k$ is minimal and positive. Furthermore, M shall be chosen sufficiently large, in order to adequately realize the underlying filter function. This is achieved by choosing M such that

$$\left|h(nT_{int} - t_{k-m})\right| < \varepsilon, \quad \forall m > M\ , \tag{7}$$

is fulfilled for a given ε which is chosen sufficiently small.

In summary, the above three requirements for the summation result in the following procedure. The computation of an actual output data value $y(nT_{int})$ starts with the multiplication of the input data value $x(t_k)$ at the time instant $t_k < nT_{int}$ lying nearest to the output time instant nT_{int} with the corresponding weight factor $w(k,n)$ which is computed according to equation (5) (showing the relation between the discrete weight factors and the continuous impulse response function $h(t)$). Then, it is checked whether the condition ($\left|h(nT_{int} - t_k)\right| < \varepsilon$ (cp. equation (7)) is fulfilled for a given ε. If this condition is not fulfilled, a further multiplication corresponding to the "subsequent" index k-1 is carried out and the result is added to the result of the multiplication corresponding to the "preceding" index k. These steps are repeated until the above condition is fulfilled. If the condition is fulfilled, the sum of all the preceding multiplica-

tions is outputted as the output data value $y(nT_{int})$.

Furthermore, in a preferred embodiment the impulse response function of a lowpass filter having a selectable cut-off frequency is used. The advantage of such an underlying filter model is that the cut-off frequency can be chosen arbitrarily to reduce undesirable aliasing effects which may occur in classical interpolation methods as described in WO 01/87647 A1 due to the following effect. Since event domain sampled signals can contain arbitrarily high frequencies, the conversion into time domain sampled data leads to a loss of frequencies when the sampling frequency of the time domain sampled data set is too low (i.e. the sample distance is too large). Merely increasing the sampling frequency however is in practice no remedy since it increases computer memory usage and computational complexity. Furthermore, all frequencies of the event domain sampled input data set above the sampling frequency are folded in the frequency range of the output data set thus leading to a deterioration of the output data set. This effect is called aliasing. Using a lowpass filter for the interpolation as described above advantageously removes the frequencies above half of the sampling frequency and thus suppresses the aliasing, if the cut-off frequency is substantially equal to half of the desired sampling frequency.

One suitable lowpass filter is for example the second order Butterworth filter, which may be formulated as:

$$H(s) = \frac{1}{1 + a_1 \frac{s}{f_0} + a_2 \left(\frac{s}{f_0}\right)^2} \quad \text{with } a_1 = \sqrt{2} \text{ and } a_2 = 1, \qquad (8)$$

wherein $H(s)$ is the frequency function of the Butterworth filter in the frequency domain, which translates into the

following impulse response in the time domain:

$$h(t)=\frac{\sqrt{2}}{f_0}e^{-\frac{f_o}{\sqrt{2}}t}\sin(\frac{f_o}{\sqrt{2}}t), \qquad t>0, \tag{9}$$

In equations (8) and (9), f_0 is the cut-off frequency of the Butterworth filter.

In another preferred embodiment, the weights of the linear interpolation are chosen according to an *Epanechnikov* filter kernel (i.e., impulse response function) (see also V.A. Epanechnikov, Non-parametric estimation of a multivariate probability density, Theory of Probability and Its Applications, 14:153-158, 1969), which may be formulated as:

$$w_d(t)=\frac{3}{4d}\left(1-\frac{t^2}{d^2}\right)_+ \tag{10}$$

wherein d denotes the size of the neighborhood (see below), and $(\cdot)_+$ denotes the positive part of the function. The Epanechnikov kernel is depicted in fig. 3 for d = 1. The size d of the neighborhood defines the number of weights used for the interpolation of the output signal value $y(nT_{int})$, and might be defined via the following surrounding I_n of the output time instant nT_{int}:

$$I_n=\left\{i\middle| nT_{int}-d<t_i<nT_{int}+d\right\} \tag{11}$$

The discretized version of the Epanechnikov kernel may be written as:

$$y(nT_{\mathrm{int}})=\frac{\sum_{i\in I_n}w_d(t_i-nT_{\mathrm{int}})x(t_i)}{\sum_{i\in I_n}w_d(t_i-nT_{\mathrm{int}})}=\sum_{i\in I_n}w(i,n)x(t_i)\ . \tag{12}$$

The effect of this particular weight factor selection is a smoothing linear interpolation of the input data set attenuating or even removing large frequencies. The parameter d can be referred to as the bandwidth of the filter, and is related to the cut-off frequency of the filter. Therefore, a possible choice for the parameter d is:

$$d = T_{int} . \qquad (13)$$

In order to reduce the computational complexity of the interpolations as described above, the weight factors of the interpolation can be pre-calculated and stored in a look-up table.

According to a further embodiment of the invention, the weight factors $w(i,n)$ of equation (4) are chosen such that the linear interpolation is a nearest neighbor linear interpolation. In the nearest neighbor linear interpolation, the $y_n = y(t_n^{out})$ for $t_{k-1} < nT_{int} < t_k$ are obtained via the formula:

$$y(nT_{int}) = \gamma\, x(t_{k-1}) + (1-\gamma)x(t_k) \quad \text{with } \gamma = \frac{t_k - nT_{int}}{t_k - t_{k-1}} . \qquad (14)$$

Here, γ and $1-\gamma$ are the weight factors for the nearest neighbor linear interpolation. Since the input data values $x(t_k)$ are event domain sampled, the weight factors γ and $1-\gamma$ will vary for every output data value $y(nT_{int})$.

When converting the input data set 12, some input data values of the input data set 12 may be not be considered in the nearest neighbor linear interpolation 14 when the sampling period of the output data set 16 is too large. The frequencies contained in the input data set 12 above the Nyquist frequency of the output data set 16 are aliased. In order to avoid this aliasing effect the nearest neighbor linear interpolation step is performed with a sampling frequency which is

a factor p times higher than the desired output sampling frequency (i.e., $1/T_{int}$). Hereby, the factor p should be chosen large enough so that the thus increased sampling frequency now comprises all frequencies contained in the input data set 12. In a subsequent step, a digital lowpass filter with a cut-off frequency equal to $1/(2T_{int})$ is applied to the result of the interpolation step. The lowpass filter eliminates all frequencies above half of the desired output sampling frequency. In a last step, the obtained intermediate data set is decimated by the factor p by keeping only every p-th element of the intermediate data set and throwing away the rest. Thus, the frequencies of the input data set 12 above half of the desired output sampling frequency are removed without aliasing effects.

In a preferred embodiment, the event domain sampled input data set 12 which is converted to the time domain sampled output data set 16 is obtained by measuring the angular motion of a rotating axle with a rotational speed sensor which is a typical example where event domain sampling occurs. The wheel axles in vehicles, the cam shaft in engines and the motor axles in robots are some examples for the application of these sensors.

A typical example for such a rotational speed sensor is shown in Fig. 4. This sensor comprises in principal an inductive transducer 30 (for example an inductive or optical transducer, a Hall sensor, etc.,) cooperating with the teeth 32 of a toothed wheel 34. Every time a tooth 32 passes the transducer 30 it generates a trigger signal. The rotational speed is determined using the time elapsed between two adjacent trigger signals. Between two adjacent trigger signals of the transducer 30, the toothed wheel 34 rotates around the angle $\alpha = 2\pi / N$, where N is the number of teeth of the wheel 34. The trigger signals are sent to a microprocessor unit consisting of an internal counter with clock period T_{clock} which

generates the event domain sampled input data set 12 according to the algorithm depicted in Fig. 5.

In an initialization step 40 a clock counter *clock* is initialized to the starting value *clock* = 0, and in a subsequent step 42 an event counter k is initialized to the starting value k = 1. After waiting until the clock period T_{clock} has passed in a step 44, the clock counter *count* is incremented in step 46. Steps 44, 46 are repeated until the next trigger signal is received. When in step 48 a next trigger signal is detected, the algorithm proceeds to step 50 where the current clock value *clock* is multiplied with the timer's clock period T_{clock} and the result is stored in t_k, the k-th input signal value of the time domain sampled input data set. The event counter k is then incremented in step 52 and the algorithm returns to step 44 waiting for a next trigger signal. The time domain sampled input data set is outputted by this algorithm in the form $(t_1, t_2, t_3, \ldots)$ indicating the time instants at which the teeth 32 pass the transducer 30. In the following, the term rotational speed sensor refers to the rotational speed sensor device shown in fig. 5 in combination with the microprocessor unit implementing the algorithm of fig. 6.

The above rotational speed sensors realize an event domain sampling as described above which can be easily seen by replacing Y_S of equation (3) by the angular distance α. The time instants t_k returned by the rotational speed sensor correspond to the angles $\alpha_k = k \cdot \alpha$ which replaces $y(t_k)$ in equation (3). In the notation of the interpolation equation (4) this is equivalent to: $t_i^{in} \mathrel{\hat{=}} t_k$ and $x(t_i^{in}) \mathrel{\hat{=}} k\alpha$. Thus, in the time interval $[t_{k-1}, t_k]$, the rotational speed ω_k of the axle to which the sensor is attached is given by $\omega_k = \alpha / (t_k - t_{k-1})$.

In practice, the toothed wheel 34 of the rotational speed

sensor is non-perfectly manufactured or may suffer wear during operation and thus shows deviations in the tooth widths and tooth distances. Fig. 6 illustrates a toothed wheel 34 wherein the outline of an ideal wheel is drawn with a dotted line and the outline of a more realistic, non-ideal wheel is drawn with a full line. The ideal tooth distance is α and δ_i is the offset error angle between tooth i and tooth i-1. Fig 7 shows, assuming that the wheel 34 is rotated with constant angular speed ω, the angular speed measured by an ideal toothed wheel sensor, which is constant in time (dotted line), and by a real, imperfectly toothed wheel sensor, which is periodically fluctuating (solid line) with a periodicity N. N is the number of teeth of the rotating wheel 32 (N = 5 in the example of fig. 7).

Fig. 8 shows the Fourier decomposition of the event domain sampled data set corresponding to the measurement of the angular velocity of a non-ideal rotational speed sensor. The periodic disturbance causes harmonics in the frequency domain with basic frequency equal to the revolution frequency $\omega_0/(2\pi)$ of the wheel 34. In fig. 8, the angular velocity is equal to $\omega_0 = 90 rad/s$ resulting in a basic frequency of 14.3 Hz. All the harmonics are multiples $f_m = m\omega_0/(2\pi)$ of the basic frequency. The harmonics may overlap with the signal of interest, and hence should preferably be removed.

In the following, a preferred embodiment for removing the harmonics which overlap with the signal of interest is described with reference to fig. 10. These harmonics are removed in a preprocessing step before the event domain sampled data set 12 of the wheel speed sensor is converted into the time domain. If for example the signal lies in the interval [30 Hz, 60 Hz], all harmonics falling in this particular interval shall be removed. For example, if the velocity is 70 rad/s, the harmonics m = 3, 4 and 5 should be removed.

The preprocessing adopted for rejection of the harmonics is based on computed angular velocity $\omega_k = \alpha/(t_k - t_{k-1})$ (calculated in step 52) between two consecutive time instants t_k of the event domain sampled data set which is obtained from the rotational speed sensor 50. The ω_k between two tooth passing events are influenced from the rotational speed of the axle, from the sensor errors and from vibrations of the wheel. By exploiting the periodicity of the sensor errors, the ω_k can be expanded using a Fourier series expansion (provided in step 54) expressing these sensor errors:

$$\omega_k = \widehat{\omega}_k^0 + \frac{1}{\Delta t_k} \sum_m a_m \cos(2\pi km/N) + b_m \sin(2\pi\, km/N) \qquad (15)$$

where $\widehat{\omega}_k^0 = 2\pi/\left(t_{k+N/2} - t_{k-N/2}\right)$ is the hypothetical angular velocity between two tooth passing events corrected for sensor errors (exploiting that $\sum \delta_i = 0$). The Fourier coefficients a_m and b_m correspond to the harmonics f_m. In order to estimate the Fourier coefficients a_m and b_m a fitting procedure, preferably a Least Mean Squares method 56 may be used. With the estimated values $\widehat{a}_m$ and $\widehat{b}_m$ it is then possible to subtract in step 58 the periodic sensor errors from the signal ω_k:

$$\widehat{\omega}_k = \omega_k - \sum_m \hat{a}_m \cos(2\pi km/N) + \hat{b}_m \sin(2\pi\, km/N)\ . \qquad (16)$$

In order to reduce the computational effort, it is possible to restrict the preprocessing to the Fourier coefficients m of interest, e.g. the harmonics which overlap with the signal of interest. This is achieved by restricting the sum in equations (15) and (16) to the values m of interest.

Fig. 9 compares the Fourier decomposition of a wheel speed

sensor's output in the frequency range from 0 *Hz* to 150 *Hz* with and without the above preprocessing. The dotted line represents the original signal calculated from the measured values ω_k. The solid line represents $\hat{\omega}_k$, the sensor error corrected signal with suppression of the sensor error induced harmonics m = 3, 4 and 5. As can be easily seen from this comparison, only the specified frequencies for these harmonics are removed from the signal. Naturally, the particular preprocessing of sensor signals may be applied irrespective of the subsequent specific interpolation as described above.

Preferably, the above described method for converting event domain sampled data sets into time domain sampled data sets is applied to signals provided by rotational speed sensors attached to an automobile wheel axle which measure the angular velocities of a tire. The tire velocity information is used for a determination of the automobile's tire pressures. This is accomplished by analyzing the vibration modes in the tire's rotational velocity. In general, the tire can be modeled by a damped spring with a resonance frequency. The resonance frequency of a tire typically depends on the tire's air pressure. This allows the tire pressure determination by measuring vibration resonances in the tire's rotational velocity.

Wheel speed sensors as described above are used for example in anti-lock braking systems (ABS) which provide a tire-road slip control and are a frequently integrated safety feature of an automobile. Thus, the rotational wheel speed sensors of the anti-lock braking system can be used simultaneously for tire pressure monitoring and tire-road slip control.

Further details of the tire pressure determination are shown in WO 01/87647 A1, the content of which is incorporated herein by reference.

According to an exemplified embodiment of the invention, the digital signal processing system may be any machine capable of executing a sequence of instructions that specify actions to be taken by that machine for causing the machine to perform any one of the methodologies discussed above. The machine may be an application specified integrated circuit (ASIC) including a processor and a memory unit. The instructions may reside, completely or at least partially, within the memory and/or within the processor.

In particular, the digital signal processing system may be implemented in the form of a computer system within which a sequence of instructions may be executed. The computer system may then further include a video display unit, an alphanumeric input device (e.g. a keyboard), a cursor control device (e.g. a mouse), a disk drive unit. The disk drive unit includes a machine-readable medium on which is stored the sequence of instructions (i.e., a computer program or software) embodying any one, or all, of the methodologies described above.

The computer program product may be a machine-readable medium which is capable of storing or encoding the sequence of instructions for execution by the machine and that cause the machine to perform any one of the methodologies of the present invention. The machine-readable medium shall accordingly be taken to include, but not be limited to, solid-state memories, optical and magnetic disks, and carrier wave signals.

In the above description, for simplification, it is only referred to the method case. The system case can be easily derived from the latter one by replacing at the appropriate places in the description the expression 'step' for the method case by the expression 'component' for the system case.

Furthermore, all other publications and existing systems mentioned in this specification are herein incorporated by reference.

Although certain methods, systems and products constructed in accordance with the teachings of the invention have been described herein, the scope of coverage of this patent is not limited thereto. On the contrary, this patent covers all embodiments of the teachings of the invention fairly falling within the scope of the appended claims either literally or under the doctrine of equivalents.

What is claimed is:

1. A digital signal processing method for digitally processing a discretized input signal (12) provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time, comprising:
converting the input signal (12) to a discretized output signal (16) representing the input signal (10) with discrete output signal values at discrete output time instants which are distributed equidistantly in time, wherein the conversion is carried out by a linear interpolation (14);
characterized in that the weight factors ($w(i,n)$) of the linear interpolation are derived from an impulse response function of a filter (18).

2. The method according to claim 1, wherein the linear interpolation (14) is carried out according to the following formula:

$$y\left(t_n^{out}\right) = \sum_i w(i,n)x\left(t_i^{in}\right),$$

$x\left(t_i^{in}\right)$ denoting the discrete input signal value at the discrete input time instant t_i^{in}, $y\left(t_i^{out}\right)$ denoting the discrete output signal value at the discrete output time instant t_i^{out}, and $w(i,n)$ denoting the weight factors.

3. The method according to claim 2, wherein the weight factors $w(i,n)$ are chosen according to the following formula:

$$w_h(i,n) = h\left(t_n^{out} - t_i^{in}\right)\left(t_i^{in} - t_{i-1}^{in}\right),$$

wherein h is the impulse response function of a continuous time filter (18).

4. The method according to anyone of the preceding claims, wherein the filter is a causal filter.

5. The method according to anyone of the preceding claims, wherein the filter is a lowpass filter having a selectable cut-off frequency.

6. The method of claim 5, wherein the filter is a Butterworth filter.

7. The method according to claim 5 or 6, wherein the cut-off frequency is automatically selected in dependence of the sampling frequency of the output signal (16).

8. The method according to anyone of claims 3 to 7, wherein only those of the weight factors ($w(i,n)$) are used for the interpolation (14) whose corresponding value of the impulse response function is greater than a predetermined lower limit.

9. The method of claim 2, wherein the impulse response of the filter (18) is an Epanechnikov kernel (fig. 3).

10. The method of claim 9, wherein the Epanechnikov kernel (fig. 3) is chosen according to the following formula:

$$w_d(t) = \frac{3}{4d}\left(1 - \frac{t^2}{d^2}\right)_+ \qquad (10)$$

wherein d is a filter parameter which is chosen in dependence of the sampling frequency, and $(\cdot)_+$ denotes the

positive part of the function.

11. A digital signal processing method for digitally processing a discretized input signal (12) provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time, comprising:
converting the input signal (12) to a discretized output signal (16) representing the input signal (10) with discrete output signal values at discrete output time instants which are distributed equidistantly in time, wherein the conversion is carried out by a nearest neighbor linear interpolation (14);
characterized in that
the nearest neighbor linear interpolation is performed with a sampling frequency which is p times higher than the desired output sampling frequency, wherein p is a real number and the output sampling frequency is the reciprocal of the constant differences between the discrete output time instants of the output signal;
a digital lowpass filter with a cut-off frequency equal to half of the desired output sampling frequency is applied to the output of the nearest neighbor linear interpolation; and
the filter output is decimated by said factor p.

12. The method of anyone of the preceding claims, wherein the input signal (10) is provided by a rotational speed sensor (30,32,34) which in particular measures the angular motion of an axle.

13. The method of claim 12, wherein the input signal (10) is preprocessed in order to remove periodical occurring sensor errors (δ_i) originating from non-ideal rotational speed sensors (30,32,34), by:
fitting (56) the coefficients (a_m, b_m) of a Fourier series

expansion (54) on the basis of the harmonics (fig. 8) corresponding to the periodicity of the periodical occurring sensor errors (δ_i) to the input signal (10);
subtracting (58) the Fourier series expansion (54) with the thus obtained coefficients (a_m, b_m) from the input signal (10).

14. The method of claim 12 or 13, wherein the input signal (10) corresponds to the angular motion of at least one tire of an automobile and is used for a determination of the automobile's tire pressure.

15. The method of anyone of claims 12 to 14, wherein the input signal (10) is provided by the rotational speed sensor of an anti-lock braking system of the automobile.

16. The method of claim 14 or 15, wherein the tire pressure is determined by a wheel vibration analysis.

17. The method of anyone of the preceding claims, wherein the weight factors ($w(i,n)$) of the interpolation (14) are stored in a look-up table.

18. A digital signal processing system for digitally processing a discretized input signal (12) provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time, comprising:
a conversion component for converting the input signal (12) to a discretized output signal (16) representing the input signal (10) with discrete output signal values at discrete output time instants which are distributed equidistantly in time, wherein the conversion is carried out by a linear interpolation (14);
characterized in that the weight factors ($w(i,n)$) of the

linear interpolation are derived from an impulse response function of a filter (18).

19. A computer program product including program code for carrying out a digital signal processing method, when executed on a computer system, for digitally processing a discretized input signal (12) provided in form of a plurality of discrete input signal values at corresponding discrete input time instants which are distributed non-equidistantly in time, comprising:

converting the input signal (12) to a discretized output signal (16) representing the input signal (10) with discrete output signal values at discrete output time instants which are distributed equidistantly in time, wherein the conversion is carried out by a linear interpolation (14);

characterized in that the weight factors ($w(i,n)$) of the linear interpolation are derived from an impulse response function of a filter (18).

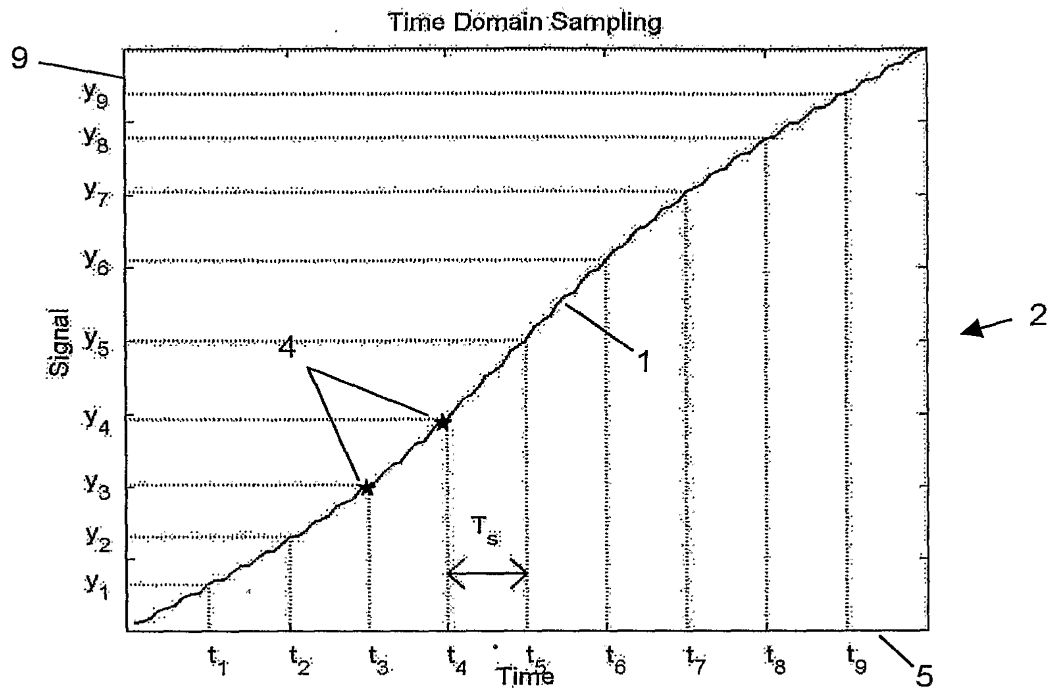
Time Domain Sampling
9
y_9
y_8
y_7
y_6
Signal
y_5
4
1
2
y_4
y_3
T_s
y_2
y_1
t_1 t_2 t_3 t_4 t_5 t_6 t_7 t_8 t_9
Time
5

Fig. 1a

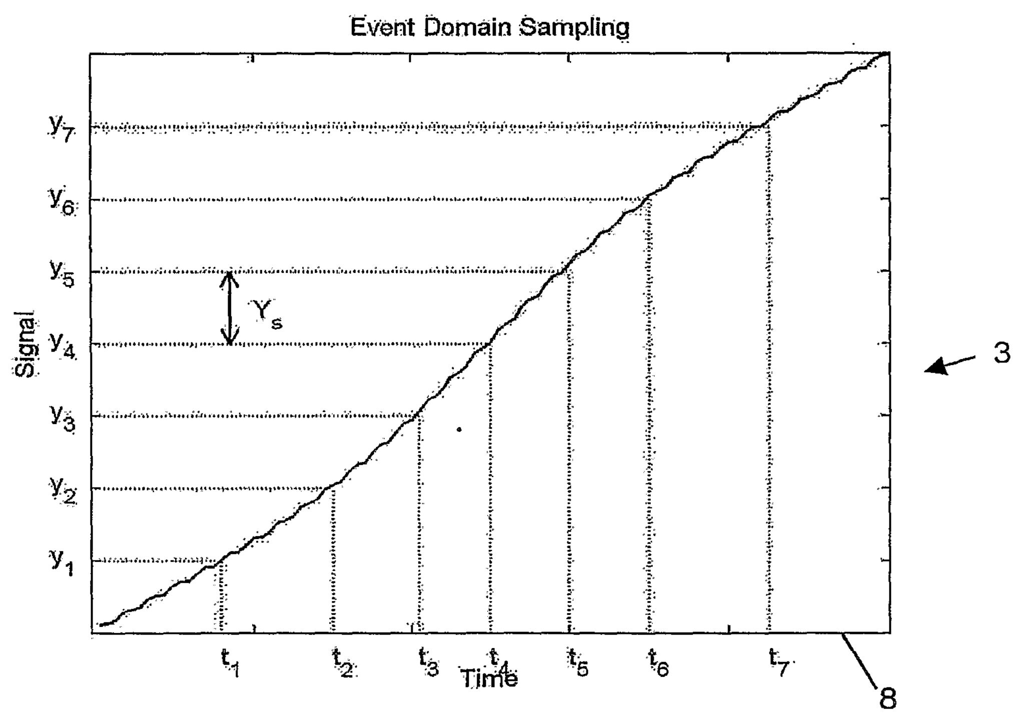
Event Domain Sampling
y_7
y_6
y_5
Y_s
Signal
y_4
3
y_3
y_2
y_1
t_1 t_2 t_3 t_4 t_5 t_6 t_7
Time
8

Fig. 1b

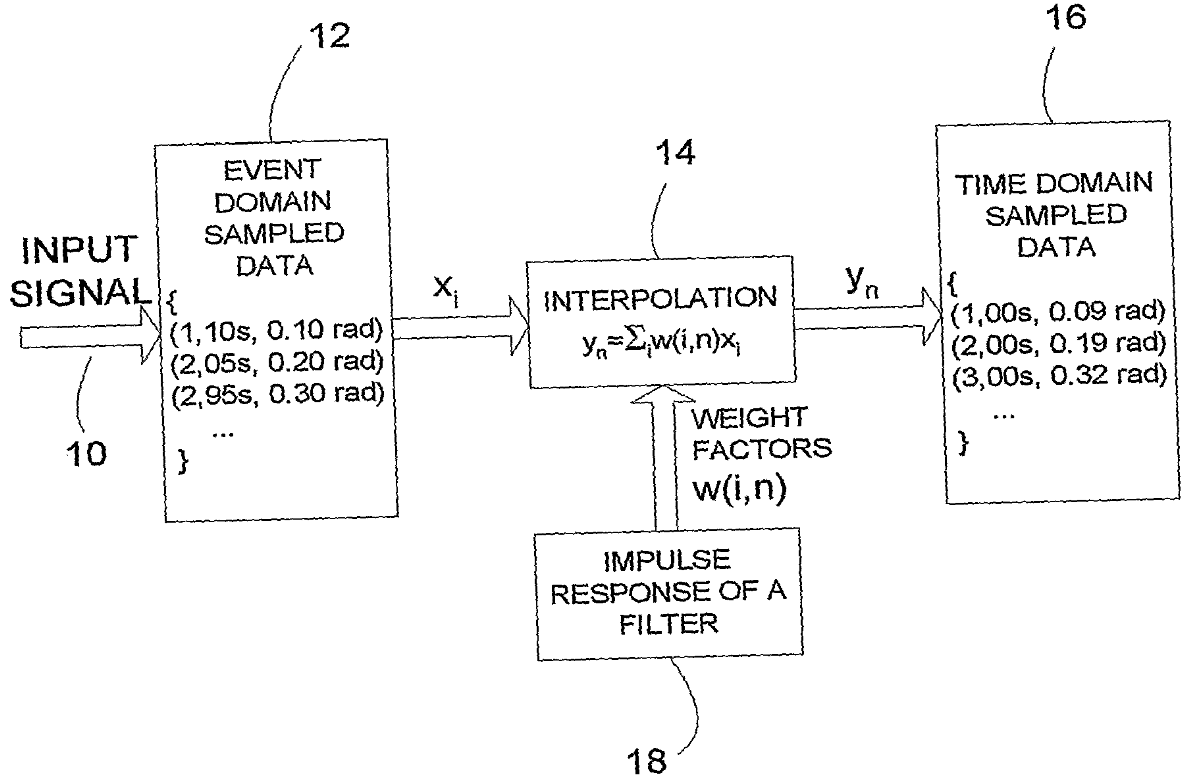
12
16
14
EVENT
DOMAIN
SAMPLED
DATA
INPUT
SIGNAL
{
(1,10s, 0.10 rad)
(2,05s, 0.20 rad)
(2,95s, 0.30 rad)
...
}
10
x_i
INTERPOLATION
$y_n = \Sigma_i w(i,n) x_i$
y_n
TIME DOMAIN
SAMPLED
DATA
{
(1,00s, 0.09 rad)
(2,00s, 0.19 rad)
(3,00s, 0.32 rad)
...
}
WEIGHT
FACTORS
w(i,n)
IMPULSE
RESPONSE OF A
FILTER
18

Fig. 2

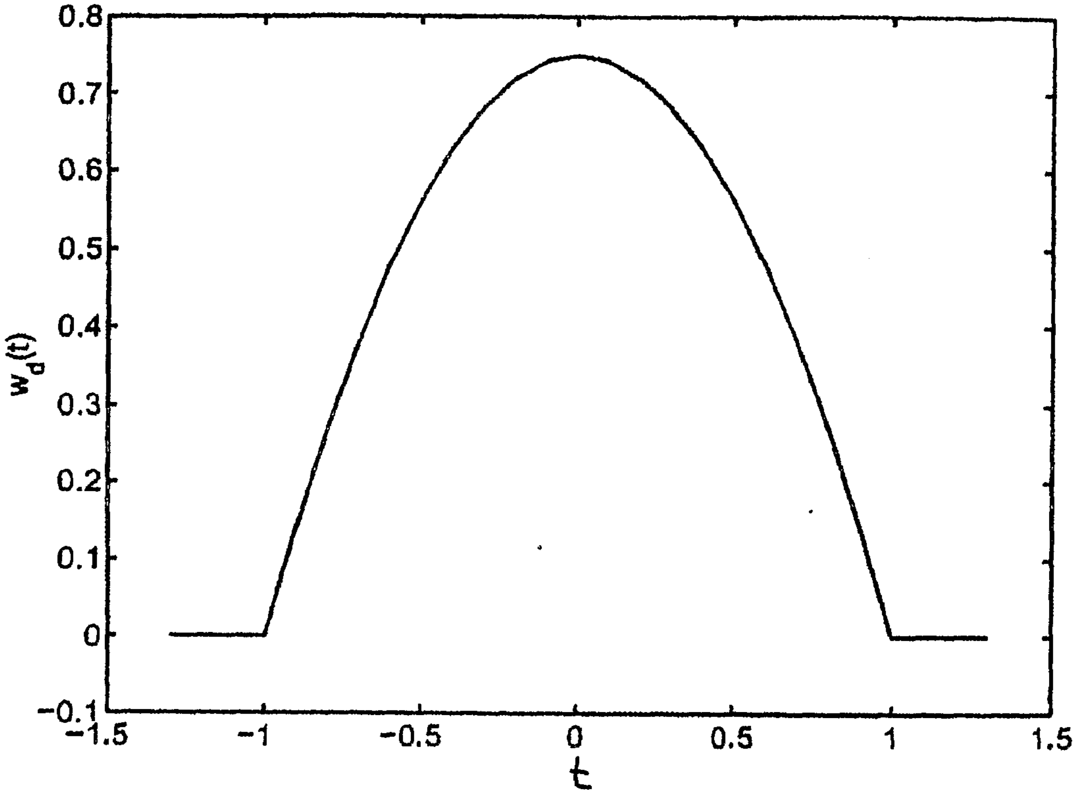

Fig. 3

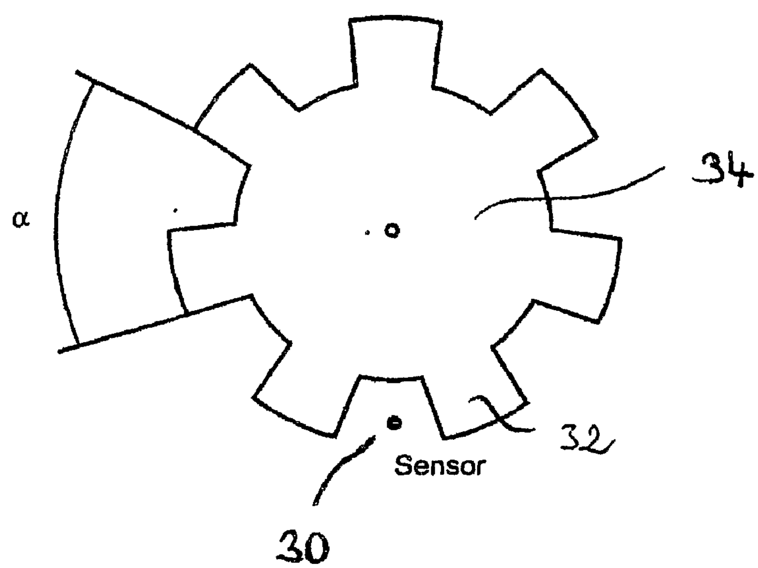

Fig. 4

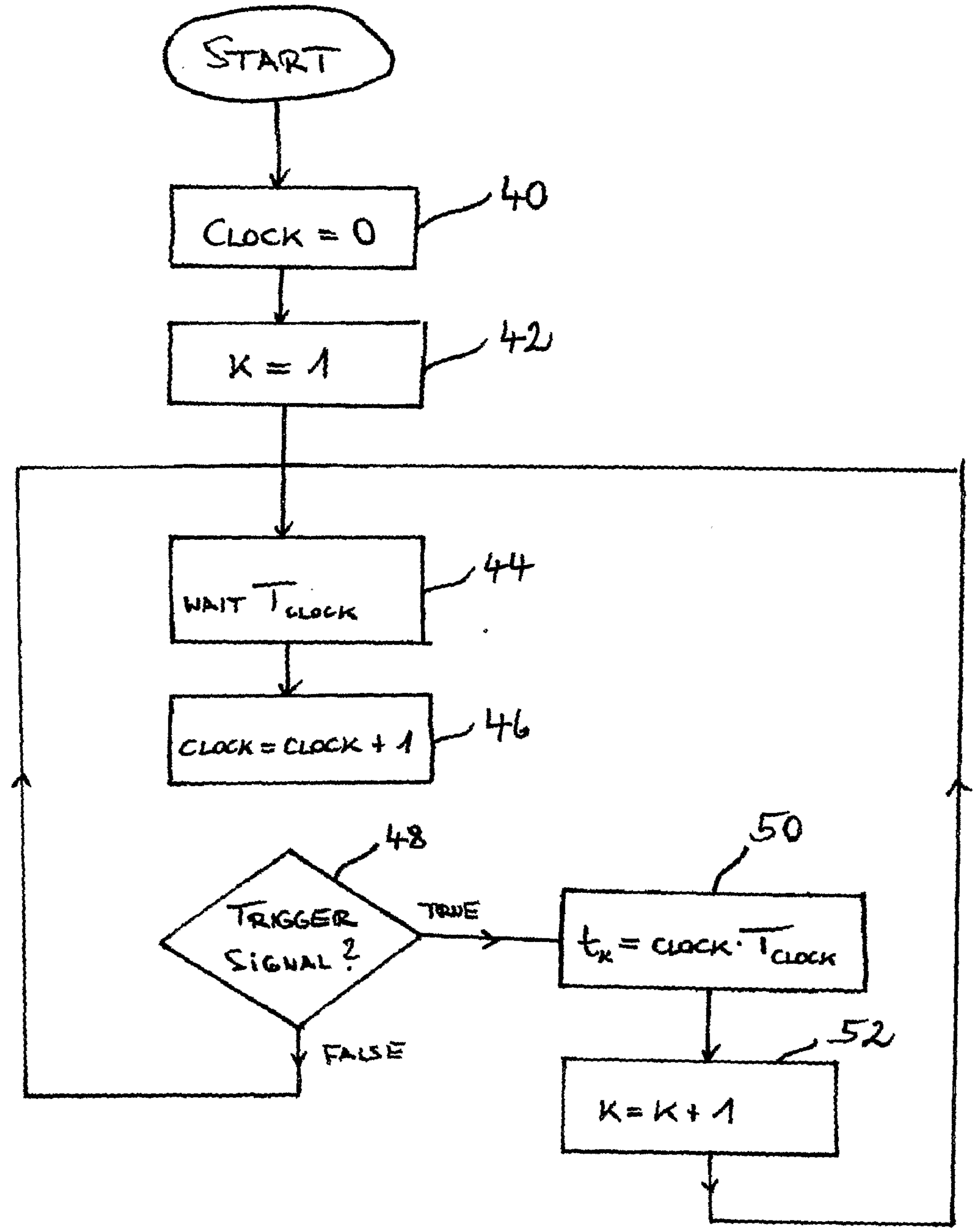
START
CLOCK = 0
40
K = 1
42
WAIT T CLOCK
44
CLOCK = CLOCK + 1
46
48
TRIGGER SIGNAL ?
TRUE
FALSE
50
t K = CLOCK · T CLOCK
52
K = K + 1

Fig. 5

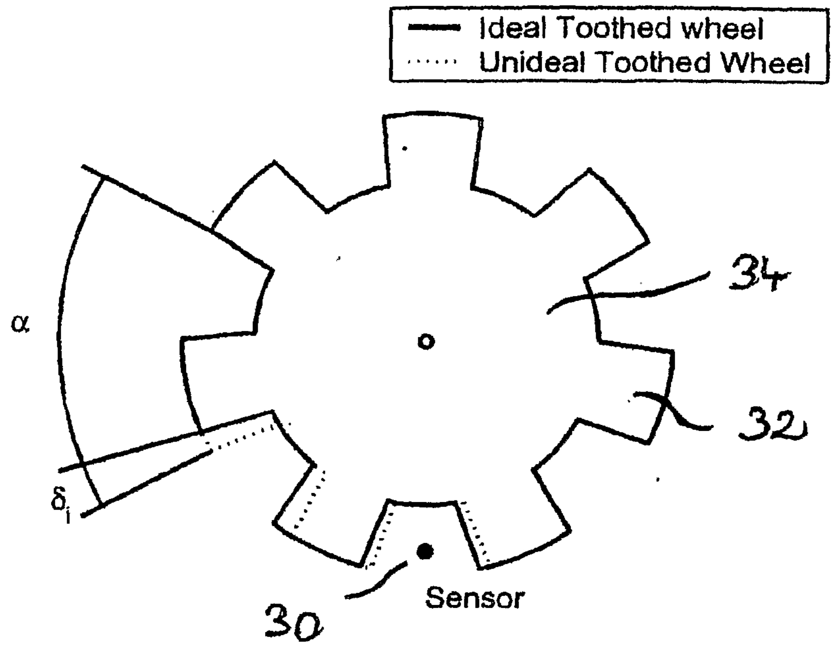
Ideal Toothed wheel
Unideal Toothed Wheel
α
δ_i
34
32
30
Sensor

Fig. 6

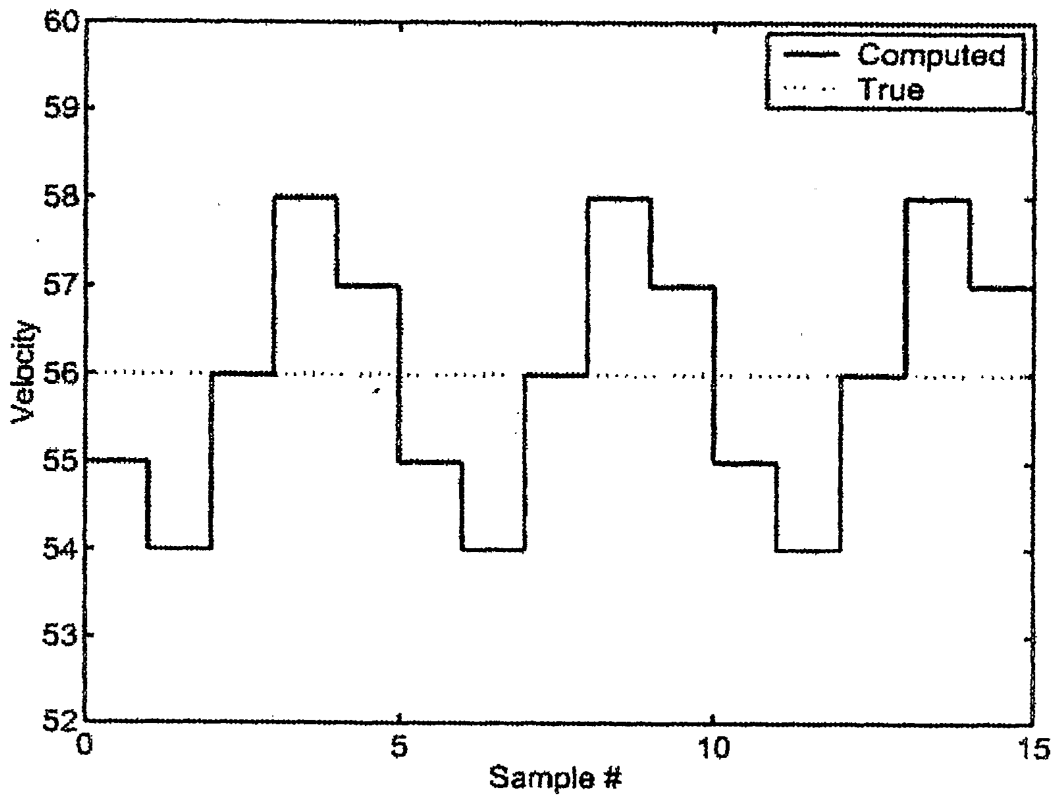
Computed
True
Velocity
Sample #
60
59
58
57
56
55
54
53
52
0
5
10
15

Fig. 7

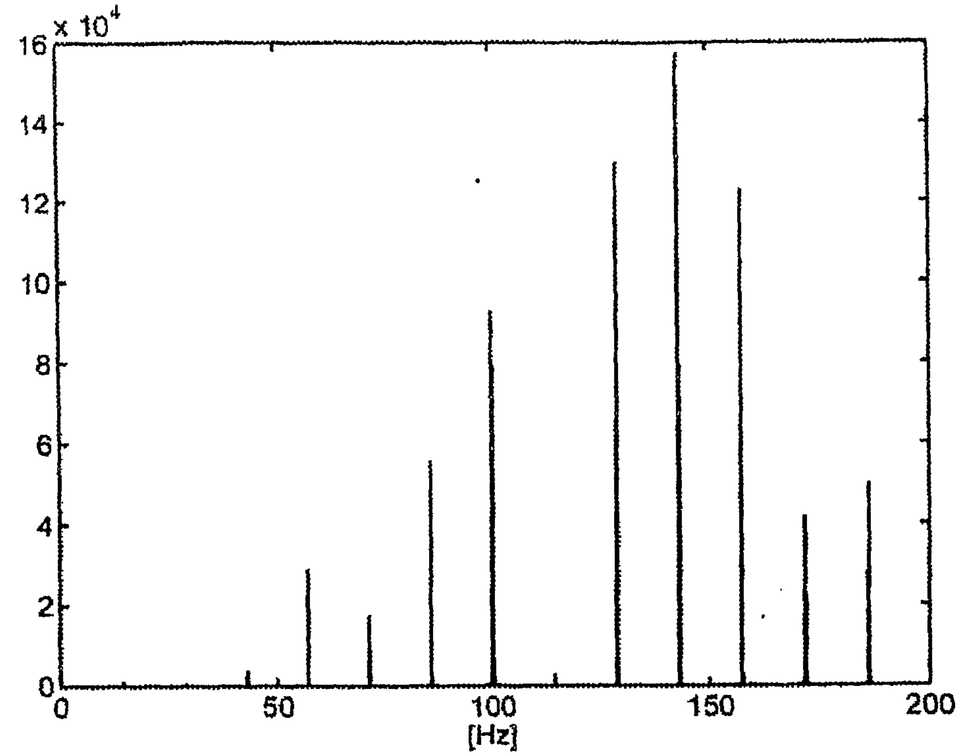
x 10^4
16
14
12
10
8
6
4
2
0
0
50
100
150
200
[Hz]

Fig. 8

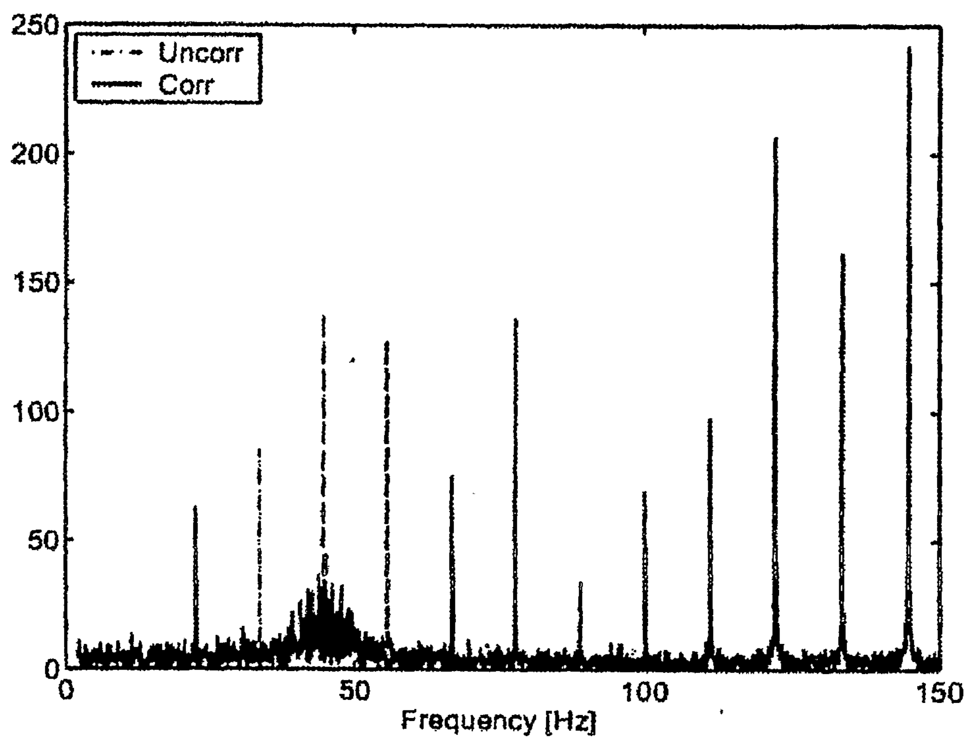
Uncorr
Corr
250
200
150
100
50
0
0
50
100
150
Frequency [Hz]

Fig. 9

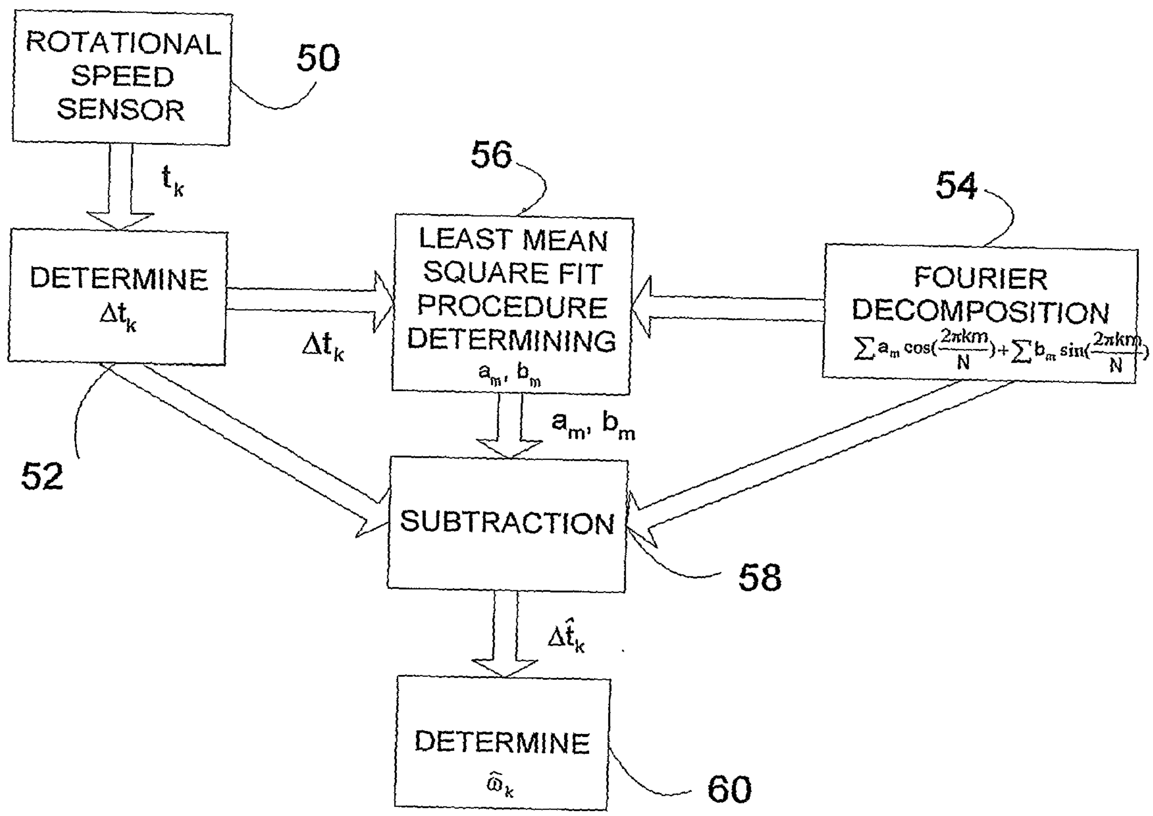
ROTATIONAL SPEED SENSOR
50
t_k
DETERMINE Δt_k
52
Δt_k
56
LEAST MEAN SQUARE FIT PROCEDURE DETERMINING a_m, b_m
54
FOURIER DECOMPOSITION
$\sum a_m \cos(\frac{2\pi km}{N}) + \sum b_m \sin(\frac{2\pi km}{N})$
a_m, b_m
SUBTRACTION
58
$\Delta\hat{t}_k$
DETERMINE $\hat{\omega}_k$
60

Fig. 10

INTERNATIONAL SEARCH REPORT

International Application No
PCT/EP 02/12409

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 G06F5/01 B60C23/06

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 G06F B60C

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, INSPEC, WPI Data, PAJ

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	WO 01 87647 A (GUSTAFSSON FREDRIK ;NIRA AUTOMOTIVE AB (SE)) 22 November 2001 (2001-11-22) cited in the application the whole document ---	1-19

-/--

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance
"E" earlier document but published on or after the international filing date
"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
15 July 2003	03. 09. 2003

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	KERSTIN WACZINSKA/MN

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	PERSSON N ET AL: "Event based sampling with application to vibration analysis in pneumatic tires" 2001 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING. PROCEEDINGS (CAT. NO.01CH37221), 2001 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING. PROCEEDINGS, SALT LAKE CITY, UT, USA, 7-11 MAY 2001, pages 3885-3888 vol.6, XP002247684 2001, Piscataway, NJ, USA, IEEE, USA ISBN: 0-7803-7041-4 the whole document ---	1-19
A	US 4 494 214 A (BERNARD FRANCIS S ET AL) 15 January 1985 (1985-01-15) the whole document -----	1-19

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/EP 02/12409

Patent document cited in search report		Publication date	Patent family member(s)			Publication date
WO 0187647	A	22-11-2001	AU	4897601	A	26-11-2001
			AU	5071501	A	23-10-2001
			EP	1272365	A1	08-01-2003
			EP	1274613	A1	15-01-2003
			SE	0002212	A	13-12-2001
			WO	0176925	A1	18-10-2001
			WO	0187647	A1	22-11-2001
US 4494214	A	15-01-1985	CA	1204170	A1	06-05-1986
			EP	0137816	A1	24-04-1985
			ES	8504395	A1	01-07-1985
			GB	2135149	A ,B	22-08-1984
			IT	1173513	B	24-06-1987
			JP	60500516	T	11-04-1985
			MY	9387	A	31-12-1987
			WO	8403159	A1	16-08-1984